Amendments to the Claims:

This listing of claims replaces all prior versions, and listings, of claims in the application.

1. (Original) A method for performing variable rate speech coding in a speech codec comprising a plurality of speech codec modes operating at different bit rates and speech encoded by said speech codec being arranged for transmission in a telecommunications network, the method comprising:

receiving information on an active codec mode set to be supported from the telecommunications network;

activating the speech-codec-supported speech codec modes that correspond to the active codec mode set determined in the telecommunications network; and

encoding speech signals to be applied to the speech codec with said activated speech codec modes such that a speech codec mode of the substantially lowest bit rate is adapted to speech frames comprised by the speech signals such that, in view of the channel conditions in the telecommunications network, the level of residual error in coding will be minimized at the same time.

2. (Original) A method as claimed in claim 1, further comprising

responsive to changes in at least either of the following: the channel conditions in the telecommunications network, the active codec mode set;

adapting the parameters to be used in the speech codec mode selection and the limit values thereof to correspond to new channel conditions and capacity of the telecommunications network or to the active codec mode set.

3. (Original) A method as claimed in claim 1, further comprising

adapting the target level of residual error in coding in the speech codec mode selection and the bit rate of the codec mode to be selected to the average bit rate employed on a traffic channel in the telecommunications network.

4. (Original) A method as claimed in claim 1, further comprising

performing at least some of the speech coding sub-processes on the speech frame; and

adapting a speech codec mode for each speech frame on the basis of the parameter values obtained from said sub-processes.

5. (Original) A method as claimed in claim 4, wherein

the speech coding is performed as ACELP coding, whereby said sub-processes include at least one of the following:

VAD parametrization process;

LPC parametrization process;

LTP parametrization process;

parametrization process of signal gain.

6. (Original) A method as claimed in claim 5, further comprising

determining the speech codec mode in two steps by

adapting a low bit rate speech codec mode for the speech frame, responsive to the parameter values obtained from the VAD parametrization process indicating that the speech frame comprises a low energy speech signal; and

adapting a higher bit rate speech codec mode for the speech frame on the basis of several said parameter values responsive to the speech codec mode of the low bit rate not being adapted for the speech frame.

7. (Original) A method as claimed in claim 4, further comprising

classifying the speech frames to be encoded into a plurality of different classes on the basis of the information analysed from the speech frames, which comprises at least some of the following: spectrum of the speech frame, gains of different speech frame parameters, zero cross frequency of the speech signal; and

adapting a speech codec mode for the speech frame on the basis of the class defined for the speech frame.

8. (Original) A variable rate speech codec comprising a plurality of speech codec modes operating at different rates and speech encoded by said speech codec being arranged for transmission in a telecommunications network, the speech codec being arranged

to receive information from the telecommunications network on an active codec mode set to be supported;

to activate the speech codec modes that correspond to the active codec mode set determined in the telecommunications network; and

to encode the speech signals to be applied to the speech codec with said activated speech codec modes such that a speech codec mode of the substantially lowest bit rate is arranged for adaption to speech frames comprised by the speech signals such that, in view of the channel conditions in the telecommunications networks, the level of residual error in coding will be minimized at the same time.

9. (Original) A speech codec as claimed in claim 8, the speech codec comprising

means for determining a speech codec mode for a speech frame from among the activated speech codec mode set by determining a speech codec mode of the substantially lowest bit rate, which mode substantially minimizes the level of residual error in coding at the same time, and

means for selecting a speech codec mode for a speech frame from among the activated speech codec mode set by adapting the level of residual error in the coding to be targeted in the speech codec mode selection and the bit rate of the selected codec mode to the average bit rate to be used on the traffic channel of the telecommunications network.

10. (Original) A speech codec as claimed in claim 9, wherein responsive to changes in at least either of the following: the channel conditions in the telecommunications network, the active codec mode set

said means for determining the speech codec mode and means for selecting the speech codec mode are arranged to adapt the parameters to be used in the speech codec mode selection and the limit values thereof to correspond to new channel conditions and capacity of the telecommunications network or to the active codec mode set.

11. (Original) A speech codec as claimed in claim 8, wherein the speech codec is arranged

to perform at least some of the sub-processes of the speech coding; and

to adapt a speech codec mode for each speech frame on the basis of the parameter values obtained from said sub-processes.

12. (Original) A speech codec as claimed in claim 11, wherein

the speech coding is arranged to be performed as ACELP coding, whereby the speech codec comprises at least one of the following:

means for performing a VAD parametrization process;

means for performing an LPC parametrization process;

means for performing an LTP parametrization process;

means for performing a signal gain parametrization process.

13. (Original) A speech codec as claimed in claim 12, wherein

the speech codec is arranged to determine the speech codec mode in two steps, whereby the speech codec comprises

means for adapting a low bit rate speech codec mode for the speech frame responsive to the parameter values obtained from the VAD parametrization process indicating that the speech frame comprises a low energy speech signal; and

means for adapting a higher bit rate speech codec mode for the speech frame on the basis of several said parameter values responsive to the speech codec mode of the low bit rate not being adapted for the speech frame.

14. (Original) A mobile station comprising a variable rate speech codec comprising a plurality of speech codec modes operating at different bit rates, the speech encoded by the speech codec being arranged for transmission in a telecommunications network, the speech codec being arranged

to receive information from the telecommunications network on the active codec mode set to be supported;

to activate the speech-codec-supported speech codec modes that correspond to the active codec mode set determined in the telecommunications network; and

to encode speech signals to be applied to the speech codec with said activated speech codec modes such that a speech codec mode of the substantially lowest bit rate is adapted to speech frames comprised by the speech signals such that, in view of the channel conditions in the telecommunications network, the level of residual error in coding will be minimized at the same time.

15. (Original) A computer program, when loaded in a processor, being arranged to implement variable rate speech codec functions, the speech codec comprising a plurality of speech codec modes operating at different bit rates, the speech encoded by the speech codec being arranged for transmission in a telecommunications network, the computer program comprising

a program code for receiving from the telecommunications network information that determines the active codec mode set to be supported;

a program code for activating the speech codec modes that correspond to the active codec mode set determined in the telecommunications network;

a program code for encoding the speech signals to be applied to the speech codec with said activated speech codec modes such that a speech codec mode of the substantially lowest bit rate is arranged for adaption for speech frames comprised by the speech signals such that, in view of the channel conditions in the telecommunications network, the level of residual error in coding will be minimized at the same time.

16. (Original) A computer program as claimed in claim 15, further comprising

a program code for determining a speech codec mode for a speech frame from among the activated speech codec mode set by determining a speech codec mode of the substantially lowest bit rate, which mode substantially minimizes the level of residual error in the coding at the same time, and

a program code for selecting a speech codec mode for a speech frame from among the activated speech codec mode set by adapting the level of residual error in the coding to be targeted in the speech codec mode selection and the bit rate of the selected codec mode to the average bit rate to be used on the traffic channel of the telecommunications network.

17. (New) A network element comprising a variable rate speech codec comprising a plurality of speech codec modes operating at different bit rates, the speech encoded by the speech codec being arranged for transmission in a telecommunications network, the speech codec being arranged

to receive information from the telecommunications network on the active codec mode set to be supported;

to activate the speech-codec-supported speech codec modes that correspond to the active codec mode set determined in the telecommunications network; and

to encode speech signals to be applied to the speech codec with said activated speech codec modes such that a speech codec mode of the substantially lowest bit rate is adapted to speech frames comprised by the speech signals such that, in view of the channel conditions in the telecommunications network, the level of residual error in coding will be minimized at the same time.

18. (New) A network element according to claim 17 which is a transcoder unit.